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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/346,884	07/02/1999	NIRAT BHUPESH SHAH	14013-23	3005

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EXAMINER
LY, ANH VU H

ART UNIT	PAPER NUMBER
2662	9

DATE MAILED: 04/24/2003

Please find below and/or attached an Office communication concerning this application or proceeding.

PTO-90C (Rev. 07-01)

Office Action Summary	Application No.	Applicant(s)	
	09/346,884	SHAH, NIRAT BHUPESH	
	Examiner	Art Unit	
	Anh-Vu H Ly	2662	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133).
- Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 29 January 2003 .

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-20 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-20 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

11)☒ The proposed drawing correction filed on 29 January 2003 is: a)☒ approved b)☐ disapproved by the Examiner.

If approved, corrected drawings are required in reply to this Office action.

12)☐ The oath or declaration is objected to by the Examiner.

Priority under 35 U.S.C. §§ 119 and 120

13)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____ .

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

14)☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. § 119(e) (to a provisional application).

a) ☐ The translation of the foreign language provisional application has been received.

15)☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. §§ 120 and/or 121.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO-1449) Paper No(s) _____ .
4) ☐ Interview Summary (PTO-413) Paper No(s). _____ .
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: .

DETAILED ACTION

Response to Amendment

1. This communication is in response to applicant's amendment filed January 29, 2003. The proposed amendment to the claims has been entered. Claims 1-20 are pending.

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claims 1-13 and 17-19 are rejected under 35 U.S.C. 103(a) as being unpatentable over Vargo et al (US Patent No. 6,356,545) in view of Blomfield-Brown et al (US Patent No. 6,292,840). Hereinafter, referred to as Vargo and Blomfield-Brown.

With respect to claims 1, 9, and 17, the limitations recited in claim 1, a DSP module responsive to an analog signal from one of the telephone devices and operative to convert analog telephone signal to digital telephone signal and further operative to packetize digital telephone signal for transmission to a remotely-located router device, are inherent to ~~by~~ Vargo.

Vargo discloses in Fig. 1, an operation of the Internet telephone system (communication system). Wherein, a call is initiated in North America (first telephone device) over a PSTN gateway server 10a (router device) from a PSTN 11a over the Internet 17 (packet switching network) to Japan and Taiwan (second telephone device).

Further, according to Fig. 1, for a call to take place over the Internet (packet switching network), the received analog signal (analog telephone signal), from the call initiator (first

telephone device) to the call receiver in Japan (second telephone device), must be digitized into packets (converted analog signal to digital signal) and transferred to the router (remotely-located router device) for routing packets through the packet network. Besides, all involved processes stated above are well known in the art in VoIP transmissions.

Further, Vargo discloses (col. 10, lines 46-67 and Fig. 11a) assuming the voice port begins with the commercially available TrueSpeech codec algorithm (first type of codec), which encodes speech at 8.5kbits/sec and with no redundancy.

After noticing dropped packets (detection of degradation in the quality of the voice information), the voice port adjusts by selecting (switching codec type) the Voxware 2.9kbits/sec algorithm (second type of codec) having somewhat lower sound quality (while conversation is taking place), but with two level redundancy error correction.

The limitations "DSP module for renegotiating the use of a second type of codec and the type of codec being utilized is repeatedly renegotiated to dynamically change compression techniques to adjust for the network usage thereby optimizing the use of network capacity and throughput" are inherent ~~by~~ to Vargo. Vargo discloses a system of dynamically selecting another type of codec once it recognizes dropped packets. As noticed in the above rejections, the voice port adjusts by selecting the Voxware 2.9kbits/sec algorithm to replace TrueSpeech codec algorithm once it notices dropped packets. Therefore, if the selected compression algorithm falls below a certain level, a new codec is chosen (DSP module for renegotiating the use of a second type of codec and the type of codec being utilized is repeatedly renegotiated to dynamically change compression techniques to adjust for the network usage thereby optimizing the use of network capacity and throughput).

Vargo does not disclose the router device and the remotely-located device initially negotiating to utilize a first type of codec.

Blomfield-Brown discloses (col. 11, line 47-56) a list of default compression schemes are maintained by the local and remote application. The local application sends a single message to the remote application requesting one of the default compression schemes. The remote application sends a message back to the local application including the default compression scheme which should be used (router device and the remotely-located device initially negotiating to utilize a first type of codec).

It would have been obvious to one having ordinary skill in the art at the time the invention was made to include the feature of initially negotiating a compression scheme between the sending end and receiving end in Vargo's system, as suggested by Blomfield-Brown, to reduce latency by reducing the number of compression negotiation messages sent.

With respect to claims 2-4, Vargo disclose (col. 4, lines 48-51) that the teleport is designed to be able to switch codecs between one data packet and the next in the same data stream.

Vargo does not disclose that wherein switching between the codes is initiated by a user of one of the telephone devices and wherein a predetermined code is assigned to each codec, the user specifies the type of codec to be switched to by entering the predetermined code corresponding to a desired codec into one of user telephone devices and predetermined code is programmably-alterable.

However, switching initiated by a user and predetermined code are well known in the art such a TV remote controller, wherein a user can select different channels to view and wherein the remote controller can be programmed to store a number of channels with associated "hot keys". Wherein, each "hot key" is corresponded with a channel and a user can press that "hot key" to turn to that specific channel. User can re-program the remote controller to different "hot keys" associated with different channels at another time.

It would have been obvious to one having ordinary skill in the art at the time the invention was made to include a method of user initiating and assigned predetermined code, which is re-programmable, for each codec in Vargo's system, to increase system's functionalities.

With respect to claim 5, Vargo discloses (col. 11, lines 18-19) that voice port 61 responds to changing network conditions (detecting lower bandwidth available on the packet switching network) to maintain speech quality. Further Vargo discloses (col. 11, lines, 20-22) that it is possible to vary the size of the individual packets or to vary the bundling-of the packets (switching from a codec resulting in the use of larger packet sizes to a codec resulting in smaller packet sizes) by techniques that are well known in the art.

With respect to claims 6 and 8, the limitation "wherein the router device automatically detecting the lower and higher bandwidth" is addressed in the rejection of claim 5. Wherein, Vargo discloses that voice port 61 responds to changing network conditions to maintain speech quality.

With respect to claim 7, Vargo discloses (col. 11, lines 18-19) that voice port 61 responds to changing network conditions (detecting higher bandwidth available on the packet switching network) to maintain speech quality. Further Vargo discloses (col. 11, lines, 20-22) that it is possible to vary the size of the individual packets or to vary the bundling-of the packets (switching from a codec resulting in the use of smaller packet sizes to a codec resulting in higher packet sizes) by techniques that are well known in the art.

With respect to claim 10, the limitation recited in claim 10 is addressed in the rejection of parent claim 1. Wherein, Vargo discloses that after noticing dropped packets (loss of one or more packets), the voice port adjusts by selecting the Voxware 2.9kbits/sec algorithm having somewhat lower sound quality, but with two level redundancy error correction.

With respect to claim 11, the limitation recited in claim 11 is addressed in the rejection of claim 10. Wherein Vargo discloses that after noticing dropped packets (threshold defines the number of lost packets), the voice port adjusts by selecting the Voxware 2.9kbits/sec algorithm having somewhat lower sound quality, but with two level redundancy error correction.

With respect to claim 12, the limitation "wherein a plurality of packets form a message and each packet includes a sequence number indicative of the position of the packet with respect to other packets in the same message, the sequence numbers of the same message being in sequential order" is addressed in the rejection of parent claim 1. Wherein Vargo discloses that a

stream of voice data 200 includes a plurality of data packets numbered 1 through 10, where each packet further contains a plurality of data bytes indicated by the letters in Fig. 8(a) to 8(d).

The limitation, wherein a loss of packets is detected when one or more sequence numbers are missing from the received packets of the same message, is addressed in rejection of claim 11. Wherein, Vargo discloses that after noticing dropped packets, the voice port adjusts by selecting the Voxware 2.9kbits/sec algorithm having somewhat lower sound quality, but with two level redundancy error correction.

With respect to claim 13, Vargo discloses (col. 1, lines 40-43) that since Internet is built to transfer data packets rather than continuous streams of sound, there may be delays and losses. Further, Vargo discloses that voice port 61 responds to changing network conditions (degradation in the quality of the voice information is due to an intolerable delay associated with the time for a packet to travel between the router device and the remotely-located router device) to maintain speech quality.

With respect to claim 18, Vargo discloses an Internet telephone system with dynamically varying codec. Vargo does not disclose codec negotiation is performed pursuant to H.245 protocol. However, H.245 protocol is well known in the art. Therefore, it would have been obvious to one having ordinary skill in the art at the time the invention was made to include such protocol in the negotiation process in Vargo's system, to be compatible with other currently existing devices.

With respect to claim 19, Vargo discloses an Internet telephone system with dynamically varying codec. Vargo does not disclose first type of codec utilizes a compression/decompression algorithm defined by any one of the standards: G.711, G726, G729 or G723.1 and second type of codec utilizes a compression/decompression algorithm defined by any one of the standards: G.711, G726, G729 or G723.1. Blomfield-Brown discloses (col. 6, line 61 – col. 7, line 1) that in the present invention, TrueSpeech audio codec is used, however, any audio codec known in the art such as GSM 6.10 Audio Codec, CCITT G.711 A-Law and u-Law codec, ADPCM codec, etc ....can be used in place of TrueSpeech. It would have been obvious to one having ordinary skill in the art at the time the invention was made to utilize any one of the compression/decompression in Vargo's system, as suggested by Blomfield-Brown, to be compatible with other devices.

3. Claims 14-16 and 20 are rejected under 35 U.S.C. 103(a) as being unpatentable over Schuster et al (US Patent No. 6,483,600) in view of Blomfield-Brown (US Patent No. 5,625,678). Hereinafter, referred to as Schuster and Blomfield-Brown.

With respect to claim 14, Schuster discloses in Fig. 1a, a data network facsimile system for transmission digitized facsimile signals from facsimile device 20 to facsimile device 80 over the packet switching network (the DSP module responsive to analog fax signals and operative to convert analog fax signals to digital fax signals and to packetize the digital fax signals for transmission through the packet switching network, to the second fax machine).

Schuster discloses (col. 9, line 36-53 and Fig. 2) internal architecture for the data network gateway 30 and 70 for use in a number of different types of applications such as Internet access,

Internet telephony, facsimile transmissions, etc .... including telephone interfaces 34a-c, fax/voice modem 40a-c, and a data network interface 46 which contains software and hardware modules to perform call routing, modem configuration, and other features (a DSP module for carrying a user-initiated telephone conversation on a telephone line connecting the first telephone device and the second telephone device through the packet switching network).

Schuster does not disclose fax transmission from the first fax machine to the second fax machine takes place on the telephone line causing a temporary interruption to the telephone conversation thereby avoiding the need for telephone connection to be disconnected prior to the fax transmission.

Blomfield-Brown discloses (col. 2, line 15-21) that when a person wants to send data (fax transmission) to the other person on a call, the sending modem temporarily mutes the handset and sends a signal directing the receiving modem to switch to data mode. When the receiving modem receives the signal, it mutes the handset and prepares to receive data. After transferring the data, both modems unmute their handsets and normal conversation ensues.

It would have been obvious to one having ordinary skill in the art at the time the invention was made to include the feature of temporary muting the telephone conversation, sending and receiving data while the conversation is on hold in Schuster's system, as suggested by Blomfield-Brown, to allow multiple applications such as voice and data running and sharing at the same time to increase the productivity and maximize the usage of such system.

With respect to claim 15, the limitation recited in claim 15 is addressed in the rejection of parent claim 14; Wherein, Blomfield-Brown discloses that when a person wants to send data to

the other person on a call, the sending modem temporarily mutes the handset and sends a signal (a fax overlay is transferred between the router device and the remotely-located prior prior to transmission of fax information) directing the receiving modem to switch to data mode. When the receiving modem receives the signal, it mutes the handset and prepares to receive data. After transferring the data, both modems unmute their handsets and normal conversation ensues.

With respect to claim 16, the limitation recited in claim 16 is addressed in the rejection of parent claim 14; Wherein, Blomfield-Brown discloses that when a person wants to send data to the other person on a call, the sending modem temporarily mutes the handset and sends a signal (the router detects a fax tone prior to transmission of the fax information) directing the receiving modem to switch to data mode. When the receiving modem receives the signal, it mutes the handset and prepares to receive data. After transferring the data, both modems unmute their handsets and normal conversation ensues (upon completion of the fax transmission the router device resumes the telephone conversation).

With respect to claim 20, Schuster discloses (col. 5, line 6-35) H.225 protocol is used in communications (connections are established pursuant to H.225 protocol).

Response to Arguments

4. Applicant's arguments filed 01/29/2003 have been fully considered but they are not persuasive.

Applicant argues on page 5, line 8-17, that Vargo et al (US Patent No. 6,356,545) fails to teach DSP module responsive to an analog signal from one of the telephone devices ... and further operative to packetize digital telephone signal for transmission to a remotely-located router device.

Examiner respectfully traverses such argument. Vargo et al disclose an Internet telephone system with dynamically varying codec. Therefore, to be able to communicate between telephone users over the Internet, digital signal processors (DSP) are needed to convert such analog telephone signal into digital signal and further packetize a number of digitized packets for transmission over the Internet. Such digital signal processors (DSP) and its efficiency for executing such operations are known in the art.

Applicant argues on page 5, line 18-22, newly added claimed limitation, the type of codec being utilized is repeated renegotiated to dynamically change compression techniques to adjust for the network usage thereby optimizing the use of network capacity and throughput, in the amended independent claims 1 and 17 are not either taught by Vargo et al or by Blomfield-Brown et al.

Examiner respectfully traverses such argument. Such claimed limitation is inherent ~~by~~ to Vargo. Wherein, Vargo discloses a system of dynamically selecting another type of codec once it recognizes dropped packets. As noticed in the rejections of independent claims 1 and 17, the voice port adjusts by selecting the Voxware 2.9kbits/sec algorithm to replace TrueSpeech codec algorithm once it notices dropped packets. Therefore, if the selected compression algorithm falls below a certain level, a new codec is chosen.

Applicant argues on page 6, line 10-17, that newly added claimed limitation "for causing transmission of voice conversations" ~~does~~ is not taught by Schuster et al, as claimed in the amended independent claim 14. Examiner respectfully traverses such argument, as noticed in the above rejection of independent claim 14, Schuster et al disclose (col. 9, line 36-53 and Fig. 2) an internal architecture for the data network gateway 30 and 70 for use in a number of different types of applications such as Internet access, Internet telephony (a router device for use in a communication system having a first telephone device for causing transmission of voice conversation and carrying voice conversation), facsimile transmissions, etc .... including telephone interfaces 34a-c, fax/voice modem 40a-c, and a data network interface 46 which contains software and hardware modules to perform call routing, modem configuration, and other features.

Conclusion

5. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Anh-Vu H Ly whose telephone number is 703-306-5675. The examiner can normally be reached on Monday-Friday 7:00am - 4:00pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Hassan Kizou can be reached on 703-305-4744. The fax phone numbers for the organization where this application or proceeding is assigned are 703-872-9314 for regular communications and 703-872-9314 for After Final communications.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the receptionist whose telephone number is 703-305-4750.

av
April 16, 2003

HASSAN KIZOU
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